Amendments to the Specification:

Please replace the paragraph beginning at page 2, line 12, with the following amended paragraph:

Wireless communications [[has]]have emerged to become a huge market as millions of people world-wide buy cellular handsets, subscribe to Personal Communications Services (PCS), and make calls on a daily basis. There are many competing technologies in the wireless communications field. Initially, cellular transmissions were made according to traditional analog radio frequency (RF) technology. But as wireless digital technology improved, it became clear that digital applications were far superior to that of analog. The three dominant wireless digital technologies existing today include Global System of Mobile communications (GSM), Time Division Multiple Access (TDMA), and Code Division Multiple Access (CDMA). Of these three digital wireless technologies, CDMA is gaining widespread popularity because of its many advantages.

Please replace the paragraph beginning at page 6, line 3, with the following amended paragraph:

The present invention pertains to an apparatus and method for using a linear predictive coding filter for the purpose of removing periodic and/or quasi-periodic interference from spread spectrum signals. Data is encoded and transmitted as a spread spectrum signal. Periodic and quasi-periodic signals which act as interference in the spectrum of interest [[is]] are effectively filtered out by using a linear predictive coding filter. The LPC filter takes a digitized received spread spectrum signal and generates a set of predictive coefficients and a set of error coefficients. The predictive coefficients represent the interfering periodic and/or quasi-periodic signals. As such, this set of predictive coefficients are discarded. The remaining error coefficients represent what is left over and thereby ~~contains~~ contain the useful transmitted data found within the spread spectrum signal. The error coefficients are used by the signal processing block to

extract the transmitted data. By implementing the LPC filter in such a novel and unique way, the present invention dramatically improves the range and data rate by which spread spectrum systems may operate.

Please replace the paragraph beginning at page 7, line 17, with the following amended paragraph:

Figure 4 shows a feedback loop with[[with]] a FIR filter.

Please replace the paragraph beginning at page 8, line 13, with the following amended paragraph:

Figure 1 shows an exemplary baseband direct sequence, spread spectrum CDMA receiver upon which the present invention may be practiced. The CDMA receiver uses Walsh functions and a separate pseudo-random code . The RF signal is received over antenna 101. The signal is then amplified by an amplifier 102. In the currently preferred embodiment, amplifier 102 is comprised of two or more baseband video amplifiers coupled in series. This can provide a ~~gainbandwidth~~ gain bandwidth product in the Terahertz range. Next, the analog signal is converted into an equivalent digital signal by an analog-to-digital converter 103. A linear predictive coding filter 104 is then used to filter out the periodic and quasi-periodic interference in the spectrum of interest. Thereupon, the signal is demodulated by multiplying it with the synchronized pseudo-random number 105. This is the same pseudo-random number associated with the transmitting base station. The signal is multiplied by a synchronized Walsh function 106 in order to eliminate interference due to other users' transmission within that cell. An integration 107 is followed by a sample and hold 108 function. Optionally, a bandpass filter is used to filter out the AM radio signals (e.g., 0.5 MHz to 1.5 MHz). Furthermore, a high pass filter may used to filter out the higher frequencies (e.g., above 30 MHz). Alternatively, notch filter(s) may be used to filter out the known interference signal(s) in the restricted bands.

Please replace the paragraph beginning at page 9, line 9, with the following amended paragraph:

It should be noted that the present invention works with any type of spread spectrum technique and at any frequency. The present invention can be implemented for cell phone, data transfers, peer-to-peer communications, satellite, military, commercial, civilian, IEEE 802.11(b), Bluetooth, as well as a wide range of different wireless transmissions schemes, formats, and medium. One such spread spectrum system is described in detail in the patent application entitled, "A Baseband Direct Sequence Spread Spectrum Transceiver," filed January 26, 2001, Serial No. 09/772,110, now U.S. Patent No. 6,982,945, and which is incorporated by reference in its entirety herein. Another spread spectrum system is described in the patent application entitled, "Application of a Pseudo-Randomly Shuffled Hadamard Function In A Wireless CDMA System," filed December 5, 2000, Serial No. 09/730,697, now U.S. Patent No. 6,829,289, and which is incorporated by reference in its entirety herein.

Please replace the paragraph beginning at page 11, line 1, with the following amended paragraph:

According to the publication, in LPC, the current input sample x(n) is approximated by a linear combination of past samples of the input signal. The prediction of x(n) is computed using an FIR filter by

$$\hat{x}(n) = \sum_{k=1}^{p} a_k x(n-k)$$

where p is the prediction order and [[a k]]$\underline{a_k}$ are the prediction coefficients. With the z-transform of the prediction filter

$$P(z) = \sum_{k=1}^{p} a_k z^{-k}$$

the difference between the original input signal x(n) and its prediction ^x(n) is evaluated in the z-domain by

$$e(z) = X(z) - \hat{X}(z) = X(z)[1 - P(z)] = X(z)A(z).$$

Please replace the paragraph beginning at page 11, line 14, with the following amended paragraph:

The difference signal e(n) is called residual or prediction error and its calculation is depicted in Figure 3. Here the ~~feed forward~~ feed-forward prediction is considered where the prediction is calculated in the forward direction from the input signal.

Using the excitation ~e(n) as input to the all-pole filter

$$H(z) = \frac{1}{1 - P(z)}$$

produces the output signal

$$Y(z) = \tilde{E}(z) \cdot H(z)$$

where H(z) can be realized with the FIR filter P (z) in a feedback loop as shown in Figure 4. If the residual e(n) calculated in the analysis stage is fed directly into the synthesis filter, the input signal x(n) will be ideally recovered.

Please replace the paragraph beginning at page 12, line 7, with the following amended paragraph:

The IIR filter H(z) is termed a synthesis filter or LPC filter and it ~~models—except for a gain factor—the~~ models—except for a gain factor—the input signal X(z). For speech coding this filter models the time-varying vocal tract. The filter A(z) = 1 - H(z) for calculating the residual from the input signal is called the inverse filter.

Please replace the paragraph beginning at page 12, line 12, with the following amended paragraph:

With optimal filter coefficients, the residual energy is minimized. This can be exploited for efficient coding of the input signal where the quantized residual ~e(n) = Q{e(n)} is used as excitation to the LPC filter. Three commonly used linear prediction methods (autocorrelation method, Burg algorithm, and gradient adaptive lattice) are presented to get methods which are suited for real-time computation, i.e. to get a similar workload each sample. We consider here linear prediction methods for a computation of the residual with zero delay. Thus, the prediction coefficients are computed from past samples of the input signal and the methods are suited for audio coding using the ADPCM structure where no transmission of the filter coefficients is required. The fast filter update coming from the similar workload of each sample leads to better spectral models than block-based approaches where the coefficients are held constant for the duration of one block. With a fast update of the spectral model no interpolation of the filter coefficients between frames is required.

Please replace the paragraph beginning at page 13, line 4 with the following amended paragraph:

The prediction error can be calculated by a standard FIR structure which requires the direct FIR coefficients a_k. In the lattice structure of Figure 2, the signals f_m (n) and b_m(n) are used which are the forward and backward prediction errors of an m-th order predictor. Forward prediction means the prediction ~~from~~ is based on past samples while in backward prediction, a sample is predicted from future samples. In the lattice methods, the lattice or PARCOR (partial correlation) coefficients k_i are calculated instead of the direct FIR coefficients. Although it is possible to calculate the direct coefficients from the lattice ones, sometimes it is useful to perform the filter operation in the lattice structure. The

main advantage of the lattice coefficients is that the stability of the LPC filter is guaranteed for $|k_i| < 1$. Furthermore, if a predictor of order m-1 is already known, for a predictor of order m only the coefficient k_m has to be calculated. For the direct FIR coefficients normally the complete coefficient set has to be changed in this case.

Please replace the paragraph beginning at page 13, line 19 with the following amended paragraph:

In case of using the lattice structure, the lattice states bi(n)have to be recalculated if the lattice coefficients [[k i]] $\underline{k_i}$ are changed. This problem does not occur in the direct FIR structure since the filter states are equal to past samples of the input signal and they are independent of the used coefficient set.

Please replace the paragraph beginning at page 13, line 25 with the following amended paragraph:

Some methods to calculate the prediction coefficients for minimizing the residual energy [[is]]are now described. First, the standard block-based approaches of the Burg Algorithm and the autocorrelation method are summarized. Then the sample based gradient adaptive lattice (GAL) method is described. Finally, modifications of the block-based methods for a sample-based calculation in a real-time system is described.

Please replace the paragraph beginning at page 14, line 7, with the following amended paragraph:

The Burg algorithm is based on the lattice structure and it minimizes for a predictor of order m in a block of length N the sum of the energies of the forward prediction error $f_m(n)$ and of the backward prediction error $b_m(n)$.

The initialization of the forward and backward prediction errors of order zero for the considered block is obtained by

$$f_0(n) = x(n), n = 0,\ldots,N-1$$
$$b_0(n) = x(n), n = 0,\ldots,N-1$$

where n denotes the time index in the considered block. For m = 1, . . . , p the following operations are performed:

Calculation of the m-th lattice coefficient

$$\left[\left[k_m = \frac{2\sum_{n=m}^{N-1}[f_{m-1}(n)b_{m-1}(n-1)}{\sum_{n=m}^{N-1}[f_{m-1}^2(n)+b_{m-1}^2(n-1)} \right]\right]$$

$$\underline{k_m = \frac{2\sum_{n=m}^{N-1}[f_{m-1}(n)b_{m-1}(n-1)]}{\sum_{n=m}^{N-1}[f_{m-1}^2(n)+b_{m-1}^2(n-1)]}}$$

Recursive calculation of the forward and backward ~~pre-diction~~ prediction errors of order m :

$$f_m = f_{m-1}(n) - k_m b_{m-1}(n-1),$$
$$n = m+1,\ldots,N-1$$
$$b_m(n) = b_{m-1}(n-1) - k_m f_{m-1}(n),$$
$$n = m,\ldots,N-1$$

Please replace the paragraph beginning at page 15, line 6, with the following amended paragraph:

The autocorrelation method minimizes the prediction error [[e(n),or]] e(n), or in terms of the lattice structure, the forward prediction error. For a block of length N an approximation of the autocorrelation sequence is calculated by

$$R(i) = \frac{1}{N}\sum_{n=i}^{N-1} u(n)u(n-i)$$

where u(n) = x(n)* w(n) is a windowed version of the considered block x(n), n = 0, . . . , N - 1. Normally, a Hamming window is used. For a predictor of order p the filter coefficients [[a i]]$\underline{a_i}$ for i = 1, . . . , p are obtained by solving the normal equations

$$\sum_{k=1}^{p} a_k R(i-k) = R(i), i = 1,\ldots,p.$$

An efficient solution of the normal equations is performed by the Levinson-Durbin recursion. First the energy of the predictor of order zero is initialized to E 0 = R(0). Afterwards the following operations are performed for m = 1, . . . , p, where ak(m) denotes the k-th coefficient of an m-th order predictor.

$$k_m = \frac{R(m) - \Sigma_{k=1}^{m-1} a_k^{(m-1)} R(m-k)}{E_{m-1}}$$
$$a_m^{(m)} = k_m$$
$$a_k^{(m)} = a_k^{(m-1)} - k_m a_{m-k}^{(m-1)}, k = 1,\ldots,m-1$$
$$E_m = (1-k_m^2)E_{m-1}$$

Please replace the paragraph beginning at page 17, line 1 with the following amended paragraph:

The simplest approach is to choose the [[µm]] $\underline{\mu_m}$ values constant. Simulations have shown that the optimum value of µ (equal for all orders for simplicity) depends highly on the used signals, the optimum value varies approximately in the range from 1 to 10. Better results are expected for gradient weights which are adaptively dependent on the expectation value of the sum of

the forward and backward prediction error energies. An approximation of this expectation value can be recursively calculated by

$$D_m(n) = \lambda D_m(n-1) + [f_{m-1}^2(n) + b_{m-1}^2(n-1)]$$

where 0<λ<1 influences the weight of older samples. The gradient weights are obtained by

$$2\mu_m = \frac{\alpha}{D_m(n)}$$

with a constant value αwhich is normally chosen to $\alpha = 1-\lambda$ for a recursive formulation of the Burg algorithm.

Please replace the paragraph beginning at page 17, line 19 with the following amended paragraph:

Both the autocorrelation method and the Burg algorithm require first an initialization process before the prediction coefficients are computed recursively. The real-time computation of a coefficient set of order p is spread over p + 1 samples; one sample for the initialization and one sample each for the p coefficients. Thus, with the counter i = 0, 1, . . . , p, 0, 1, . . . ~~(changing every sample)we~~ (changing every sample) we get the following procedure:

For ~~i= =0~~ i=0 perform the initialization
For i∈{1,p} calculate the coefficient with index i.

In the initialization process (i = 0) the operations are performed for setting both the forward and backward prediction error of order zero to the input samples x(n) in the considered block. For i = 1, . . . , p one coefficient k_i is calculated and applying the recursions of order m = i, the forward and backward prediction errors of i-th order are computed which are required in the following sample for computing k_{i+1}. The new k_i

replaces the previously used k_i. Since one coefficient is changed, a recalculation of the lattice states prior to the filter operation is required.

Please replace the paragraph beginning at page 18, line 23, with the following amended paragraph:

Table [[2]]1shows the maximum workloads per sample for calculating the filter coefficients, i.e. the filter operations to calculate the prediction are not considered. In the GAL the prediction order p has the greatest influence on the complexity. In this method for each sample p divisions are required which are very expensive on a DSP (but division is no required if μ is fixed). Note that in the Burg algorithm the maximum workload is only influenced by the block length N which is also the case in the autocorrelation method for long blocks.

Method	DSP instructions
suc. Autocorrelation	Max{4N+55,3N+5p+172}
suc. Burg	9N+72
GAL	90p+1

Table [[2]]1: Maximum workload per sample for coefficient calculation.

Please replace the paragraph beginning at page 19, line 10, with the following amended paragraph:

In the example given above, it can be seen that the resulting linear predictive terms are given by k_m. In the prior art, the k_m terms are used to model speech. However, note that there are resulting error terms,~~ e(n)=fp(n) and bp(n)~~ $e(n)=f_p(n)$ and $b_p(n)$, which are also generated as an output from the LPC filter. In the prior art, these error terms are simply discarded and not used in any way. However, in the present invention, the error term is kept whereas the linear predictive terms are discarded. In other words, the

~~e(n)=fp(n), bp(n)~~ $e(n)=f_p(n)$, $b_p(n)$ signals (or any linear combination thereof) are passed on through the receiver, while the k_m signal is discarded.

Please replace the paragraph beginning at page 21, line 14, with the following amended paragraph:

It should be noted that the present invention can be applied to any modulation scheme, either wireless or hard-wired, which utilizes a spread-spectrum technique. In particular, the present invention can be applied to CDMA cellular handsets as well as other wireless mobile CDMA devices or appliances. Furthermore, the present invention can be expeditiously applied to peer-to-peer wireless applications, especially for peer-to-peer cellular voice and/or data communications. The present invention can be applied to increase the range and/or data rates of existing spread spectrum techniques, including but not limited to CDMA, IEEE 802.11(b), Bluetooth, 3G, etc. One such system upon which the present invention may be practiced is described in the patent application entitled, "A Communications Network Quality Of Service System And Method For Real Time Information," filed December 12, 2000, Serial Number 09/738,010, now U.S. Patent No. 7,142,536, which is incorporated by reference in its entirety herein. In addition, the present invention works with any type of filter which has the property of discriminating periodic and/or quasi-periodic signals, including but not limited to LPC filters, adaptive gradient lattice filters, etc.